What is claimed is:

1. An internet telephone communication system comprising:

a voice receiving part receiving a first set of voice data packets through an internet network and sending a retransmission frequency information packet requesting to retransmit a same set of voice data packets R times, R being a retransmission frequency and being determined based on a data loss rate of said first set of voice data packets received; and

a voice transmitting part retransmitting said same set of voice data packets R times through an internet network according to said retransmission frequency information packet received.

2. The internet telephone communication system of claim 1, wherein said data loss rate is calculated by:

$$L = (T - D)/T$$

and

$$T = M - N$$

where

L is said data loss rate,

T is a number of voice data packets supposed to be received during a time interval,

D is a number of voice data packets received during said time interval,

M is a maximum sequence number of said voice data packets received during said time interval, and

N is a minimum sequence number of said voice data packets received during said time interval.

3. The internet telephone communication system of claim 2, wherein said time interval is set to 30 seconds.

4. The internet telephone communication system of claim 1, wherein each voice data packet includes a RTP protocol header region containing a corresponding packet sequence number and at least one data region.

5. The internet telephone communication system of claim 1, wherein said retransmission frequency information packet includes an IP header region, a UDP header region, a service identifier region being indicative of said retransmission frequency information packet, a session ID number region being newly assigned for each telephone call, and a retransmission frequency region.

6. The internet telephone communication system of claim 5, wherein said service identifier region, said session ID number region, and said retransmission frequency region have sizes of 4 bytes, 3 bytes, and 1 byte, respectively.

7. The internet telephone communication system of claim 6, wherein each retransmission frequency of 1, 2, 3 and 4 is represented as 0000 0001, 0000 0010, 0000 0100, and 0000 1000 in said retransmission frequency region, respectively.

8. The internet telephone communication system of claim 1, wherein said voice transmitting part comprising:

a formation time information adder for adding formation time information to compressively encoded voice data received;

a copy generator for generating R copies of said compressively encoded voice data containing said formation time information; and

a transmitting protocol processor for generating voice data packets based on said R copies generated and sending said voice data packets to said voice receiving part.

9. The internet telephone communication system of claim 1, wherein said voice receiving part comprising:

a data eliminator for leaving only one set of data among sets of compressed voice data that are repeatedly received and deleting all other data;

a data loss determiner for determining whether said only set of data left is damaged and calculating a corresponding data loss rate; and

a retransmission frequency determiner for determining a retransmission frequency based on said data loss rate and sending a retransmission frequency information packet containing said frequency to said transmitting part.

10. The internet telephone communication system of claim 9, wherein said retransmission frequency is determined by comparing said data loss rate with a first and second allowed data loss values set by a user.

11. The internet telephone communication system of claim 10, wherein said first and second allowed data loss values are 5% and 1%, respectively, and said retransmission frequency increases by one if said data loss rate is greater than 5% and decreases by one if said data loss rate is less than 1%.

12. The internet telephone communication system of claim 11, wherein the maximum and minimum of said retransmission frequency are set to 4 and 1, respectively.

13. The internet telephone communication system of claim 9, wherein said data eliminator deletes other voice data received using formation time information attached to said voice data.

14. A method of operating an internet telephone communication system having a voice transmitting part and a voice receiving part, the method comprising the steps of:

calculating a data loss rate of voice data packets received during a given time interval by said voice receiving part;

updating a retransmission frequency by increasing said frequency by one if said data loss rate is greater than a maximum allowed value and decreasing said frequency by one if said data loss rate is less than a minimum allowed value, said maximum and minimum allowed values being set by a user;

transmitting a retransmission frequency information packet to said voice transmitting part, said information packet containing said updated frequency; and

transmitting each voice data packet from said voice transmitting part to said voice receiving part R times, R being said updated frequency.

15. The method of claim 14, wherein said time interval is set to 30 seconds.

16. The method of claim 14, wherein said frequency is not updated if said data loss rate is between said maximum allowed value and said minimum allowed value.

17. The method of claim 14, wherein the maximum and minimum of said updated frequency are set to 4 and 1, respectively.

18. The method of claim 14, wherein said maximum and minimum allowed values are set to 5% and 1%, respectively.

19. The method of claim 14, wherein each of said voice data packets received includes a RTP protocol header region containing a corresponding packet sequence number and at least one data region.

20. The method of claim 14, wherein said retransmission frequency information packet includes a IP header region, a UDP header region, a 4 bytes service identifier region, a 3 bytes session ID number region, and 1 byte retransmission frequency region.

21. The method of claim 14, wherein said data loss rate is calculated by:

$$L = (T - D)/T$$

and

$$T = M - N$$

where

L is said data loss rate,

T is a number of voice data packets supposed to be received during said time interval,

D is a number of voice data packets received during said time interval,

M is a maximum sequence number of said voice data packets received during said time interval, and

N is a minimum sequence number of said voice data packets received during said time interval.